

PU030237 (JP1050655) ON 8804

(11) Patent Publication number: 64-50655

(19) Patent Agency of Japan (JP)

(12) Official Report on Patent Publication (A)

(43) Date of publication of application: 27.02.1989

(51) Int. Cl.⁴ Identification sign Patent Agency
reference number:

H 04 M 1/60

C - 7608 – 5K

H 04 B 3/23

7323 – 5K

Patent claims amount: no claim; claims paragraph
amount: 1;

(9 pages in all)

(54) Title of the invention: Loudspeaker telephone set

(21) Application number: 62-206528

(22) Date of filing: 21.08.1987

(71) Applicant: Hitachi LTD

(72) Inventor: Ota Gichu;

(72) Inventor: Ezaki Tomohiro.

Specifications

1. Title of the invention:

Loudspeaker Telephone Set

2. Field of Patent Claims:

1. A loudspeaker telephone set is provided with: a
sending signal path including a microphone for sending;
a receiving signal path including a speaker for
receiving; an echo canceling circuit for canceling an
echo signal to be mixed with an output signal of the
above mentioned microphone, studying an echo path

generated by bonding the above mentioned sending signal path with above mentioned receiving signal path; and a hearable tone signal generation means at which a hearable tone is generated from the above mentioned speaker, supplying a hearable tone signal to the above mentioned receiving signal path before telephone conversation. At a loudspeaker telephone set, a switch which opens the above mentioned sending signal path at the period when the above mentioned speaker generates the above mentioned hearable tone, and shuts the above mentioned sending signal path during telephone conversation is arranged, and the above mentioned echo canceling circuit possibly composes a study of echo path by the above mentioned hearable tone before starting a telephone conversation.

2. A loudspeaker telephone set, according to [Claim 1], has means for maintaining an echo path data obtained studying by the above mentioned echo canceling circuit by the above mentioned hearable tone, and converts the echo path data maintained by the above mentioned means into initial data of the above mentioned echo canceling circuit during telephone conversation.

3. A loudspeaker telephone set, according to [Claim 1] or [Claim 2] supplies the above mentioned hearable tone signal into the above mentioned receiving signal path through a nonlinear circuit.

4. At a loudspeaker telephone set, according to [Claim 1] or [Claim 2], the above mentioned hearable tone signal generation means consists of the first means for generating a hearable tone required by sending and receiving, and audio composing means.

5. A loudspeaker telephone set, according to [Claim 4], supplies a hearable tone signal from the above mentioned first means and above mentioned audio composing means to the above mentioned receiving signal path at the same time, before telephone conversation.

6. A loudspeaker telephone set, according to [Claim 4], supplies a hearable tone signal from only the above mentioned audio composing means to the above mentioned receiving signal path before telephone conversation.

3. Detailed explanation of the Invention:

[Technical Field of the Invention]:

The present invention relates to a suitable loudspeaker telephone set using in car telephone, conference telephone or the like, especially, it relates to a loudspeaker telephone set of echo cancelling system prevented a howling generated by sound banding, reflecting a sound from speaker on a wall, and incoming then into microphone.

[Technique of the Prior Art]:

A loudspeaker telephone set can perform a telephone conversation by a fixed speaker and microphone without arranging a handset (sender/receiver). The sending audio signal generated by input of sending audio into the microphone is sent through a hybrid circuit which performs 2 lines – 4 lines conversion or the like, after it was amplified by a sending amp; also, the receiving audio signal is supplied into the speaker, after it was amplified by a receiving amp, passing the hybrid circuit, a receiving audio is generated. Consequently, the loudspeaker telephone set can freely use both means even during telephone conversation, especially, it is useful either in a car telephone at which attention has been recently paid, for safe driving during telephone conversation, or in conference telephone for easy writing during telephone conversation.

However, in case when the loudspeaker telephone set is used in car telephone or conference telephone, the loudspeaker telephone set is arranged in car or room. Therefore, a receiving audio output from the speaker is reflected on a window glass of car or wall of room, and inputted into the microphone, and a loop at which this output signal is reflected on the hybrid circuit or the like, and then supplied to the speaker is generated.

Consequently, if the loudspeaker telephone set is constituted by the microphone, speaker and amp, a howling is generated by loop of this signal, and there is apprehension that telephone conversation will be impossible.

An audio switch system has been priority used in the loudspeaker telephone set to prevent this howling. It is a system for breaking off this loop, inputting means which generates various damages in the receiving signal path during sending, and oppositely, in the sending signal path during reception. Consequently, at this system, simultaneous telephone conversation is impossible due to damage enters some telephone conversation path. Also, a cut occurs in the beginning or end of word of conversation by change of damage, and unnatural sense is provided to conversation.

It has been recently paid attention on echo cancelling system as a system changed the above mentioned system.

A loudspeaker telephone set of this echo cancelling system arranges an echo cancelling circuit 3 for denying only a signal $y(t)$ reflected on a wall of room or the like, being outputted from a speaker 2, among signal $s(t) + y(t)$ to be inputted into a microphone 1, as shown in FIGURE 7.

After the sending signal $s(t)$ was inputted into the microphone 1, and amplified by a sending amp 4, passing the echo cancelling circuit 3, it is transmitted,

being sent from a hybrid circuit 6 into a subscribing line 7. Also, a sending signal $x(t)$ sent, passing the subscribing line 7, is supplied from the hybrid circuit 6 into the sending amp 5, and after being amplified there, is supplied into the speaker 2.

At the echo cancelling circuit 3, a transversal filter 31 is a filter having characteristic which approximates a transmission function (echo path) which is $y(t)$ by reflection of signal $x(t)$ inputted to the speaker 1 on a wall of room or the like (pseudo echo path), generally, it is a filter having an approximate tap coefficient and impulse respond of former transmission function. A sending signal $s(t)$ is outputted, denying only an echo signal $y(t)$, by deducting a pseudo echo signal $\hat{y}(t)(=y(t))$ at a subtraction device 32 from input signal $y(t)+s(t)$ of the microphone 1, creating the pseudo echo signal $y(t)$ from the signal $x(t)$ at this transversal filter 31.

The tap coefficient of this transversal filter 31 is required one after another at the tap coefficient presumption circuit 33 by a commonly known algorithm which is LMS method (Least Mean Square Method) or study identification method from signal $x(t)$ inputted to the speaker 2 and output signal $e(t)$ of the subtraction device 32. The tap coefficient presumption circuit 33 adds correlation in tap coefficient one after another, basing on input signal $e(t)$ and signal $x(t)$ input to the speaker 2 by this algorithm, it is provided to

the transversal filter 31 one after another as a tap coefficient. At the last, the tap coefficient of the transversal filter 31 becomes approximate of impulse respond of echo path, and the echo signal $y(t)$ is almost denied from input signal $s(t) + y(t)$ of the microphone 1.

A presumption of this tap coefficient should be performed when signal input to the microphone is only echo signal $y(t)$. If not, the echo signal $y(t)$ is masked by the sending signal $s(t)$, and presumption of tap coefficient is incorrect. Therefore, an electric power of input signal $s(t) + y(t)$ of the microphone 1 is compared with electric power of signal inputted to the speaker 2, namely, receiving signal $x(t)$, it is determined whether the sending signal $s(t)$ exists, when the electric power of the input signal $s(t)$ of the microphone 1 has a constant amount bigger then the electric power of the receiving signal $x(t)$, and a double talk detector 34 for prohibition an update of tap coefficient of the tap coefficient presumption circuit 33 is arranged.

Thus, loop of a former signal is broken off between the microphone 1 and the speaker 2, and howling is prevented. Also, at this system, damage should not be input in conversation path as in the audio switch system, therefore, simultaneous conversation is possible, and a good conversation quality is obtained without occurrence of cut in the beginning of word or end of word.

However, some extent of time is required for good approximating of the echo path by the transversal filter, namely, for enough denying the echo signal $y(t)$ from input signal of the microphone. It, usually, starts at the beginning, setting a value of tap coefficient to be zero, updates the tap coefficient one after another, following to the former algorithm, and at last, there is a tap coefficient which approximates an impulse respond of the echo path, therefore, it depends on number of tap coefficients but, a time of some sec from some 100 m sec is required. At this time, a deny of enough echo signal is not performed (elimination amount is small), therefore, howling can occur.

To eliminate it, a noise, chirp serial signal, reamer sine wave signal or impulse serial signal are generated from the speaker as a training signal before telephone conversation, a tap coefficient is previously required by former algorithm, operation of echo cancelling circuit is started by this tap coefficient, and telephone conversation is performed, as described in the prior techniques patent publication number of which is 58-90832, 60-117928, 61-3536.

[The problem which the Present Invention intends to solve]:

However, at the above mentioned prior technique, an especial training signal is used, therefore, unpleasant feeling of tone occurs in user by it.

However, it is usually generated before telephone conversation, therefore, feeling of user is got more and more irritated.

Also, a timing during which these training signals are sent comes off from a usual telephone operation sequence, therefore, unnatural feeling occurs in user.

The purpose of the present invention is to provide a loudspeaker telephone set capable of enough eliminating a howling, without occurring of unnatural and unpleasant feelings in user, eliminating the above mentioned problems.

[Means for solving to the Problem]:

To reach the above mentioned problem, the present invention arranges a switch for opening a sending signal path at the period of hearable tone such as a bell tone to be generated, generated audio before sending, receiving, and shutting the above mentioned sending signal path during telephone conversation, to operate an echo cancelling circuit at the period of the said hearable tone and during telephone conversation.

[Operation]:

To open a sending signal path during a hearable tone, a hearable tone generated from speaker, is inputted into microphone being reflected on wall of room or the like, and an output signal of this microphone is sent only to echo cancelling circuit.

Consequently, the echo cancelling circuit can perform a detection of characteristic of echo path, namely, a study of echo path by the above mentioned hearable tone. A result of this study is initial data, and during conversation, the above mentioned echo cancelling circuit doesn't perform study of echo path, and eliminates echo signal from output signal of the microphone. Consequently, elimination of echo signal is enough and correctly performed from start of telephone conversation, and a good telephone conversation can be performed, also, the above mentioned hearable tone is a sound to be experienced when user utilizes usual telephone set, and unpleasant sound doesn't occur.

[Description of the preferred embodiment]:

The preferred embodiment of the present invention will be described further below referring to the drawings.

FIGURE 1 is a block diagram showing the preferred embodiment of a loudspeaker telephone set according to the present invention. 8 is a switch, 9 is a dispatch tone generating circuit; 10 is an automatic hook switch, 11 is a call signal detection circuit, 12 is a bell tone generating circuit, 13 is a call tone detection circuit, 14 is a busy tone detection circuit, 15 is a mixing circuit, 16 is a tap coefficient holding circuit, 17 is a hook button switch, 18 is a push switch, 19 is a control circuit.

FIGURE 1 has the same numbers as that of parts shown in FIGURE 7, therefore, their explanation is omitted.

At FIGURE 1, the switch 8 opens and shuts between a sending amp 4 and hybrid circuit 6. The dispatch tone generating circuit 9 generates a hearable tone such as DTMF signal due to information about telephone number of partner of telephone conversation to converter (not shown) during dispatching. The automatic hook switch 10 connects the hybrid circuit 6 to a subscriber line 7. The call signal detection circuit 11 detects a call signal from the converter. The bell tone generating circuit 12 is driven by an output signal of the call signal detection circuit 11, to generate a hearable tone (bell tone) which informs user about principle of call (arrival of call). The call tone detection circuit 13 detects a call tone to be sent by converter to transmit that a dialed collocutor is during calling out, to dispatcher. A busy tone detection circuit 14 detects a busy tone to be sent by converter to inform dispatcher that the dialed collocutor is during conversation. The mixing circuit 15 electrically mixes an output signal of the bell tone generating circuit 12 and signal from the hybrid circuit 6. The tap coefficient maintaining circuit 16 temporally maintains a tap coefficient data of the transversal filter 31 which is output of the tap coefficient presumption circuit 33.

The hook button switch 17 is for informing the loudspeaker telephone set about telephone conversation wish of user. The control circuit 19 performs the whole control.

Then, operation of the preferred embodiment will be explained, but first of all, operation during arrival a mail will be explained using FIGURE 2.

If call signal is sent from the converter (not shown) through the subscriber line 7, the call signal detection circuit 11 detects this call signal to inform the control circuit 19 about principle of call (Step 201). Therefore, the control circuit 19 controls the bell tone generating circuit 12 at the same time with opening of the switch 8 (Step 202) to generate a bell tone signal by output signal of the call signal detection circuit 11. This bell tone signal is loudly spoken by the speaker 2, being amplified by the receiving amp 5. User can be informed about arrival of mail by this bell tone. Also, the control circuit 19 makes to operate the echo cancelling circuit 3 (Step 203).

If user wishes for telephone conversation, hearing a bell tone, user presses the hook button switch 17 (Step 204). If the control circuit 19 detects that the hook button switch 17 was pressed, a tap coefficient data which is output of the tap coefficient presumption circuit 33 of this point of time is recorded to the tap coefficient maintaining circuit 16 (Step 205), to stop operation of the echo cancelling circuit 3 (Step 206).

Then, it controls the bell tone generating circuit 12 to stop a loud speaking of bell tone (Step 207).

At the above mentioned operation, while a bell tone is being loudly spoken by the bell tone generating circuit 12, this bell tone is input to the microphone 1, being reflected on wall of room or the like, an output signal of this microphone 1 is supplied to the echo cancelling circuit 3. Consequently, at the echo cancelling circuit 3, this bell tone is used, the echo cancelling operation explained in FIGURE 7 is performed following to the prescribed algorithm, and a characteristic of echo path of room or the like, namely, impulse respond of room or the like is required one after another by the tap coefficient presumption circuit 33. Consequently, when the hook button switch 17 was pressed, the tap coefficient data which expresses enough a characteristic of echo path of room or the like of this point of time is stored to the tap coefficient maintaining circuit 16. At this time, the switch 8 is opened, and loop of signal is not formed, therefore, howling doesn't occur. If operation of the bell tone generating circuit 12 is stopped, the control circuit 19 supplies a tap coefficient data to be stored to the tap coefficient maintaining circuit 16 to the transversal filter 31 as initial value, to operate again the echo cancelling circuit 3 (Step 208). Then, the switch 8 is shut (Step 209), then, the automatic hook switch 10 is shut (Step 210), and the hybrid circuit 6 is connected to

the subscriber line 7. Thus, the user can process a telephone conversation with partner (Step 211). At this time, the echo cancelling circuit 3 starts operation from the state which has enough echo cancelling characteristics, therefore, even if a loop of signal is formed, the howling doesn't almost occur, and a pleasant telephone conversation can be performed from the beginning. If telephone conversation is finished by pressing the hook switch button 17 again by the user (Step 212), the control circuit 19 stops operation of the echo cancelling circuit 3 (Step 213), opens the automatic hook switch 10 (Step 214), and separates the hybrid circuit 6 from the subscriber line 7.

Then, operation during despatch will be explained further below referring to FIGURE 3.

A telephone number is inputted from the push button 18 and dispatch is performed, pressing the hook button switch 17 (Step 301), instead of picking up of handset of usual telephone set. If the control circuit 19 detects that the hook button switch 17 was pressed, it opens the switch 8 (Step 302), and starts operation of the echo cancelling circuit 3 (Step 303).

Then, the automatic hook switch 10 is shut (Step 304), and hybrid circuit 6 is connected to the subscriber line 7. If a telephone number of partner is inputted using the push button 18 by user (Step 305), the control circuit 19 detects this telephone number, generates 2 frequencies signal (DTMF signal) at which signals with

different frequencies are obtained, being mixed in every numerical value of telephone number, controlling the dispatch tone generating circuit 9, and then transmits it to the converter, sending from the hybrid circuit 6 to the subscriber line 7. This DTMF signal is also branched by the hybrid circuit 6, and then is loudly spoken by the speaker 2, passing the mixing circuit 15 and receiving amp 5 (Step 306). The user can confirm that a telephone number was inputted by this loud speaking.

Also, the audio from the speaker 2 is inputted to the microphone 1 being reflected on wall of room or the like, and output signal of the microphone is supplied to the echo cancelling circuit 3. At the echo cancelling circuit 3, an impulse respond of room is required one after another by the tap coefficient presumption circuit 33 from signal to be supplied from the dispatch tone generating circuit 9 through the hybrid circuit 6, mixing circuit 15 and receiving amp 5, and the signal from the microphone 1.

The converter connects circuit to the collocutor at the point of time when input of telephone number was finished. At this time, if collocutor is during conversation, a busy tone signal is sent to dispatch terminal (loud speaker telephone set), if collocutor is not during conversation, a call signal is sent to collocutor, and call tone signal is sent to dispatch terminal.

If the collocutor is during telephone conversation, the busy tone detection circuit 14 detects it, and inform the control circuit 19 (Step 307). Also, this busy tone passes the hybrid circuit 6, and is loudly spoken by the speaker 2, through the mixing circuit 15, and receiving amp 5 (Step 309). The user hears it, and gets to know that the collocutor is during telephone conversation, and presses the hook button 17 again (Step 310). It is the same with operation of hanging on a handset of usual telephone set. Thus, the control circuit 19 stops operation of the echo cancelling circuit 3 (Step 319), opens the automatic hook switch 10 (Step 320), and separates the hybrid circuit 6 from the subscriber line 7.

If the collocutor is not during telephone conversation, a call tone signal from the converter is detected by the call tone detection circuit 13, and the control circuit 19 is informed about result of detection (Step 308). Also, this call tone signal passes the hybrid circuit 6, and then is supplied to the speaker 2 through the mixing circuit 15, and receiving amp 5, from this moment, a call tone is loudly spoken (Step 311). The user can know about process of calling to the collocutor by the converter, by hearing this call tone. If a handset is picked up for telephone conversation by collocutor, the converter stops sending of call tone signal. The call tone detection circuit 13 detects a sending stop of this call tone signal, a call tone from the speaker 2 is eliminated, and user can get to know that the collocutor

is in handset, by this. The control circuit 19 makes to record the tap coefficient of the tap coefficient presumption circuit 33 of point of time when call tone stopped, to the tap coefficient maintaining circuit 16, by detecting a stop of this call tone by the call tone detection circuit 13 (Step 314). Then, operation of the echo cancelling circuit 3 is stopped (Step 314).

After that, the tap coefficient data to be stored to the tap coefficient maintaining circuit 16 is sent to the transversal filter 31 as initial value, to operate again the echo cancelling circuit 3 (Step 315), and to shut the switch 8 (Step 316). Thus, the user can process a telephone conversation with collocutor (Step 317). At this time, the echo cancelling circuit 3 studies enough an echo path characteristic of room, using a dispatch tone and call tone, and starts operation, converting its tap coefficient into initial value, therefore, a pleasant telephone conversation can be performed without occurrence of howling, even if loop of signal is formed. If the telephone conversation is finished, by pressing the hook button switch 17 again by the user (Step 318), the control circuit 19 stops operation of the echo cancelling circuit 3 (Step 319), opens the automatic hook switch 10 (Step 320), and separates the hybrid circuit 6 from the subscriber line 7.

As mentioned above, at the preferred embodiment, the echo cancelling circuit 3 is previously operated, as if loop of signal was broken off, using a hearable tone

required for telephone set (bell tone during mail arrival, dispatch tone during dispatch, and call tone), the impulse respond of room is required one after another by a prescribed algorithm, then stored, and entered to the telephone conversation, and operation of echo cancelling is performed, converting a previously required impulse respond into initial value, shutting a loop of signal, therefore, enough amount of echo cancelling can be obtained from the beginning during telephone conversation, and a pleasant telephone conversation can be performed from the beginning without howling.

Thus, at the preferred embodiment, a filter for imitating an echo path is a transversal filter, but it is not limited by this, for example, the other filter such as a lattice filter of a round form.

Also, at the above mentioned operation, a hearing tone finishes, and operation of the echo cancelling circuit 3 is once stopped (Step 206, Step 314), and the operation is performed again when telephone conversation starts (Step 208, Step 315), but it can be operated continuously as it was, without stoppage. In this case, the tap coefficient should not be maintained in the tap coefficient maintaining circuit 16. Also, a study during dispatching can be performed only at dispatch tone.

FIGURE 4 is a block diagram showing the other preferred embodiment of a loudspeaker telephone set according to the present invention. 20 is a selection switch, FIGURE 4 has the same numbers with that of parts shown in FIGURE 1, therefore, their explanations are omitted.

In FIGURE 4, the selection switch is controlled by the control circuit 19, to select output of the dispatch tone circuit 9 (A side) or output of the hybrid circuit 6 (B side), as input of one side of the mixing circuit 15.

At the preferred embodiment shown in FIGURE 1, the dispatch tone signal is supplied to the mixing circuit 15, branching the hybrid circuit 6, therefore, its bell is decreased to extent of 10dB, and then is loudly spoken. There is the selection switch 20 to prevent this. While an echo path of room is being studied using dispatch tone, the selection switch 20 is connected to B side, namely, a dispatch tone signal to be outputted by the dispatch tone generating circuit 9 is directly supplied to the mixing circuit 15 to prevent its level decrease. Operation of other preferred embodiment is the same with operation of the preferred embodiment shown in FIGURE 1, therefore, its explanation is omitted.

FIGURE 5 is a block diagram showing the other preferred embodiment of a loudspeaker telephone set according to the present invention. 21 is a nonlinear circuit, 22 is a selection switch.

FIGURE 5 has the same numbers with that of parts shown in FIGURE 1, therefore, their explanations are omitted.

In FIGURE 5, the selection switch 22 is shut in B side during telephone conversation, and is open in A side while the echo path of room is being studied using hearable tone. A bell tone, dispatch tone or call tone is distorted by the nonlinear circuit 21, and then it is loudly spoken being converted into signal included a lot of high/low frequency components. At each preferred embodiment explained above, a hearable tone is near to sine wave, and there is a small number of frequency components for structuring. Therefore, if a waveform of hearable tone is as it was, and an impulse respond of room is presumed, it differs a lot from a waveform during telephone conversation, therefore, it is considered that presumed degree of precision is worsen. The nonlinear circuit 21 is for preventing this, it generates a distortion, performing a process such as a clipping and slicing in a hearable tone signal. Consequently, a tone to be loudly spoken from the speaker 2 included a lot of frequency components, and presumed degree of precision of the impulse respond of room is improved.

FIGURE 6 is a block diagram showing the other preferred embodiment of a loudspeaker telephone set according to the present invention. 23 is an audio composing circuit.

FIGURE 6 has the same numbers with that of parts shown in FIGURE 1, therefore, their explanations are omitted.

As explained in the preferred embodiment shown in FIGURE 5, a bell tone, dispatch tone and call tone are not composed by a lot of frequency components, thus, the presumption of impulse respond of room can be not enough. At FIGURE 6, the audio composing circuit 23 is arranged to prevent this. While the echo path of room is being studied, for example, when a bell tone is loudly spoken, the audio composing circuit 23 is controlled by the control circuit 19, and audio signal such as [telephone is ringing] is generated. This audio signal is supplied to the mixing circuit 15 with a bell tone signal, and then it is loudly spoken with a bell tone by the speaker 2. When there is dispatch tone, an audio corresponded to number of this dispatch tone, an audio signal at which, for example, audio [one] is generated in case of [1] number, is inputted to the mixing circuit 15 to be overlapped in the dispatch tone. Thus, a result at which an input miss of telephone number directly becomes clear is obtained. In case of call tone, an audio signal at which audio [Someone is calling out now] is generated is inputted into the mixing circuit 15 to be overlapped in the call tone. Thus, a study of the echo path of room can be performed, without providing any unpleasantness to the user, by outputting a sense of required hearable tone as an audio signal, however, this

audio consists of a lot of frequency components, therefore, a presumption of impulse response of room can be performed with enough high degree of precision.

Thus, as mentioned above, an audio signal is overlapped in a hearable tone and loudly spoken, but only a loud speaking of audio signal from the audio composing circuit 23 can be performed, stopping a loud speaking of hearable tone.

The preferred embodiment of the present invention was explained above, referring to example of home loudspeaker telephone set, but the present invention is not limited by this, and it can be also used in a car telephone set. In case when the present invention is used in a car telephone set, the subscriber line 7 is wireless, a sending and receiving is distributed in independent wireless channel. Consequently, there is no correspondence to the hybrid circuit 6 shown in FIGURE 1 in the terminal of car telephone set.

However, if a general telephone is connected to a car telephone set, there is a hybrid circuit in the converter connected to general telephone, and loop of signal can be performed at this hybrid circuit with the speaker and microphone of car telephone set similarly with that shown in FIGURE 1. Also, at car telephone set, a hearable tone such as bell tone or call tone also exists as requirement for telephone communication.

[Result of the Present Invention]:

As explained above, in accordance with the present invention at the echo cancelling circuit, a study of the echo path is performed, using a hearable tone usually experienced by the user, being generated before telephone conversation, therefore, enough echo cancelling result is obtained from the beginning of telephone conversation, and a good telephone conversation can be performed, but unpleasantness doesn't occur during the above mentioned study.

4. Brief description of the drawings:

FIGURE 1 is a block diagram showing the preferred embodiment of a loudspeaker telephone set according to the present invention;

FIGURE 2 and FIGURE 3 are flow charts showing operation of the preferred embodiment of a loudspeaker telephone set according to the present invention;

FIGURE 4, FIGURE 5 and FIGURE 6 are block diagrams showing the other preferred embodiments of a loudspeaker telephone set according to the present invention;

FIGURE 7 is a block diagram of required parts showing one example of the prior loudspeaker telephone set.

[Description of Numbers]:

- 1 is a microphone;
- 2 is a speaker;
- 3 is an echo canceling circuit;
- 8 is a switch;
- 9 is a dispatch tone generating circuit;
- 11 is a call signal detection circuit;
- 12 is a bell tone generating circuit;
- 13 is a call tone detection circuit;
- 14 is a busy tone detection circuit;
- 15 is a mixing circuit;
- 16 is a tap coefficient maintaining circuit;
- 20 is a selection switch;
- 21 is a nonlinear circuit;
- 22 is a selection switch;
- 23 is an audio composing circuit.

FIG. 1

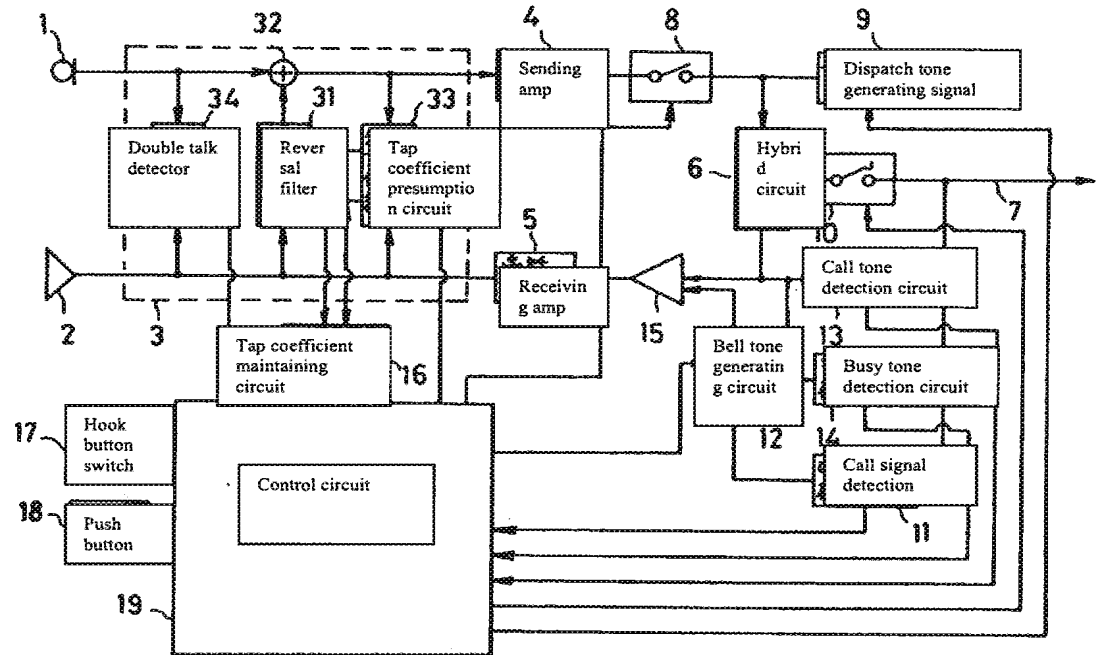


FIG.2

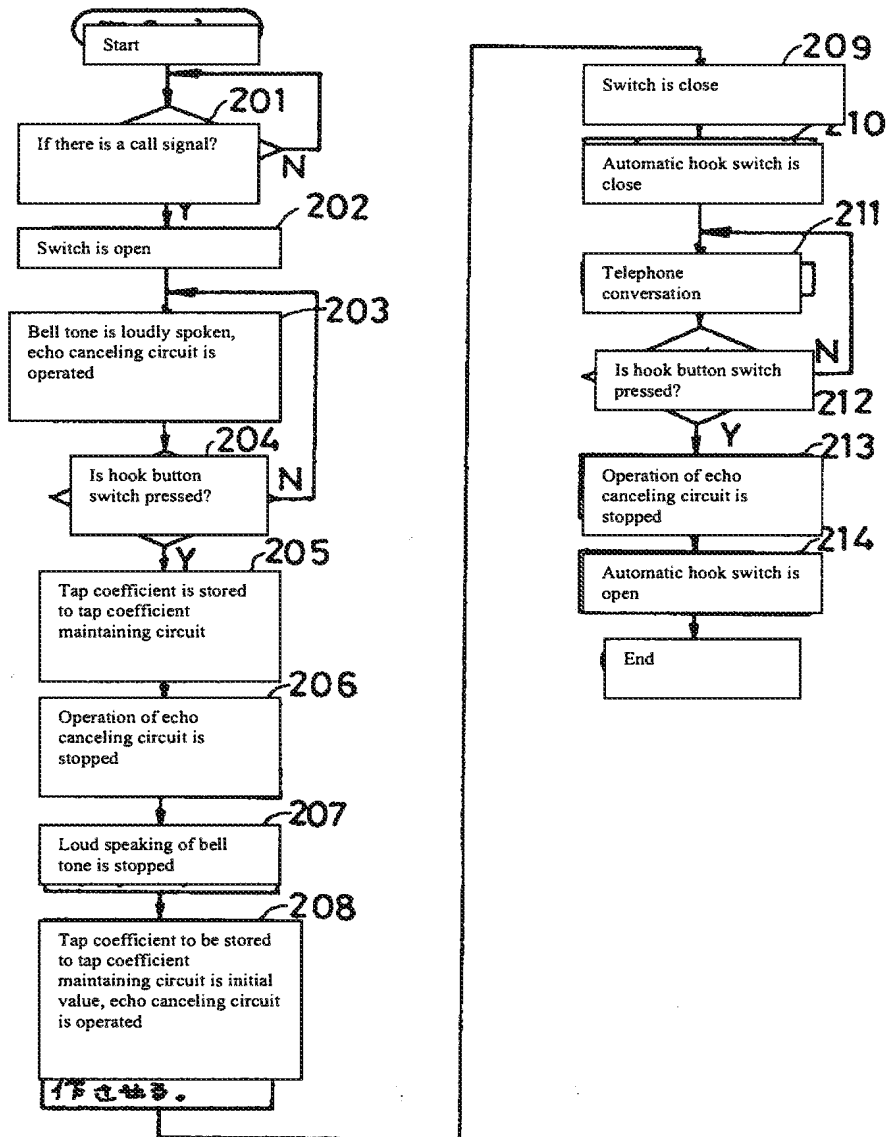


FIG.3

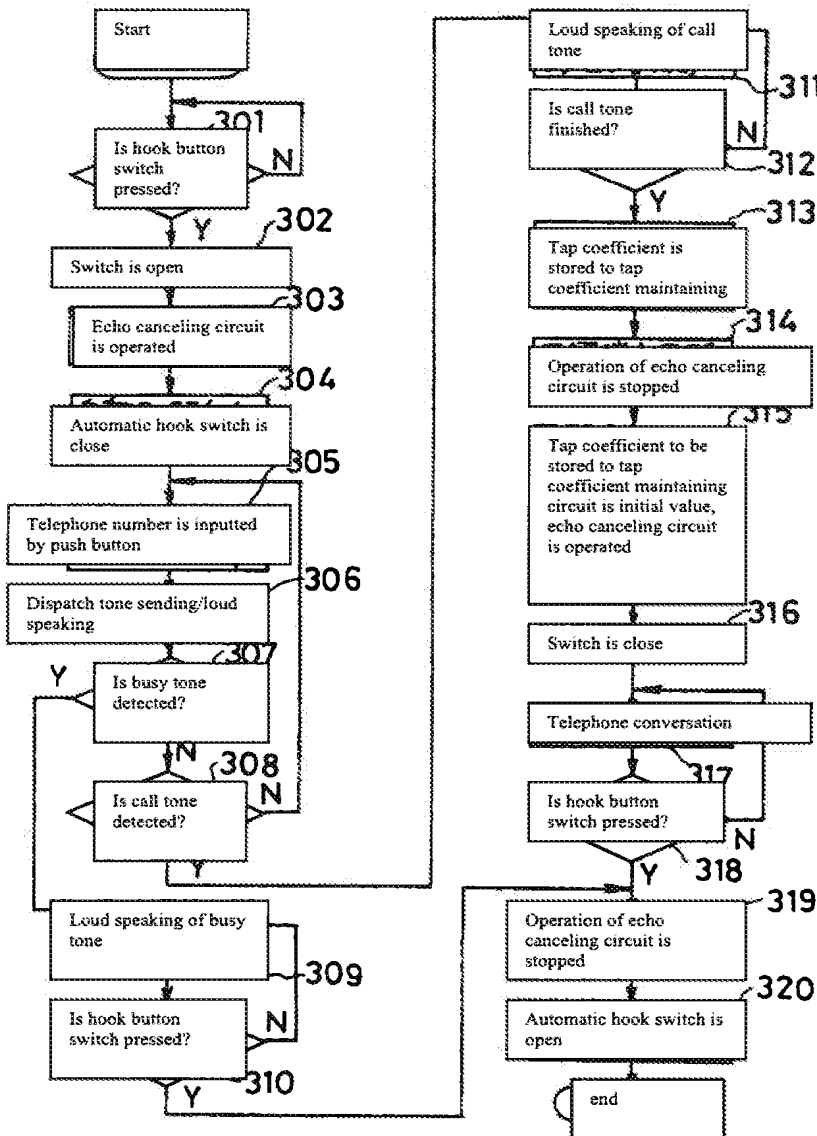


FIG.4

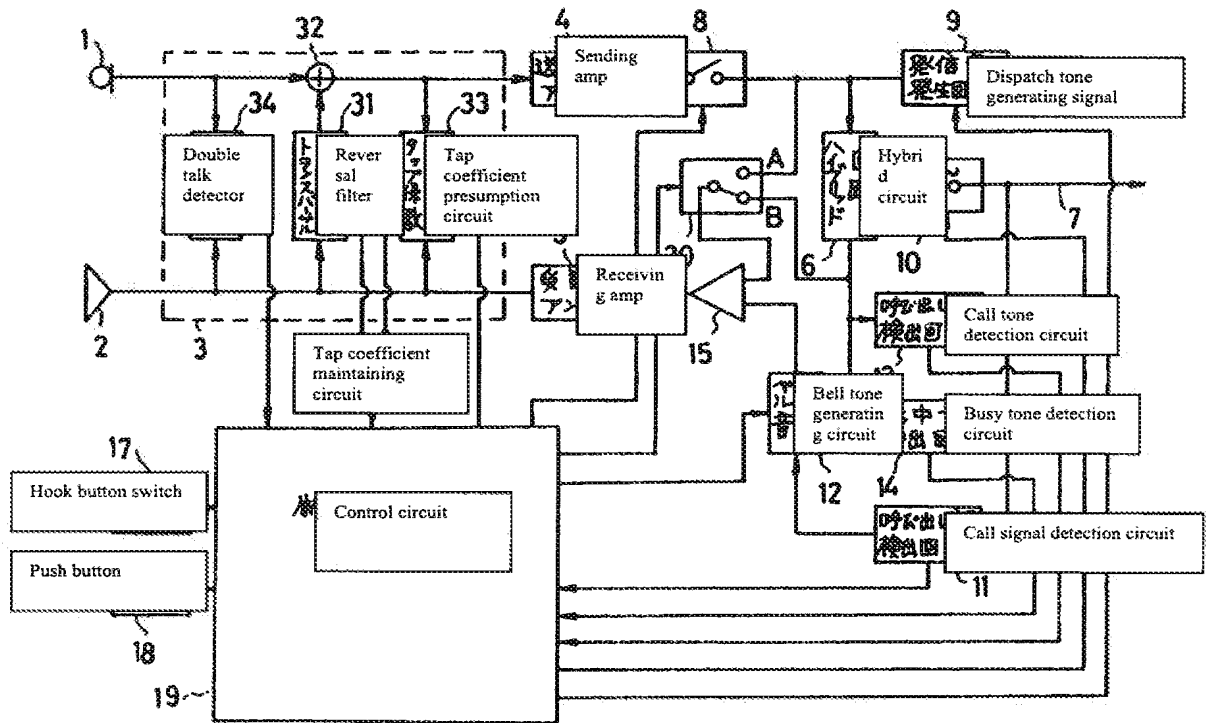


FIG.5

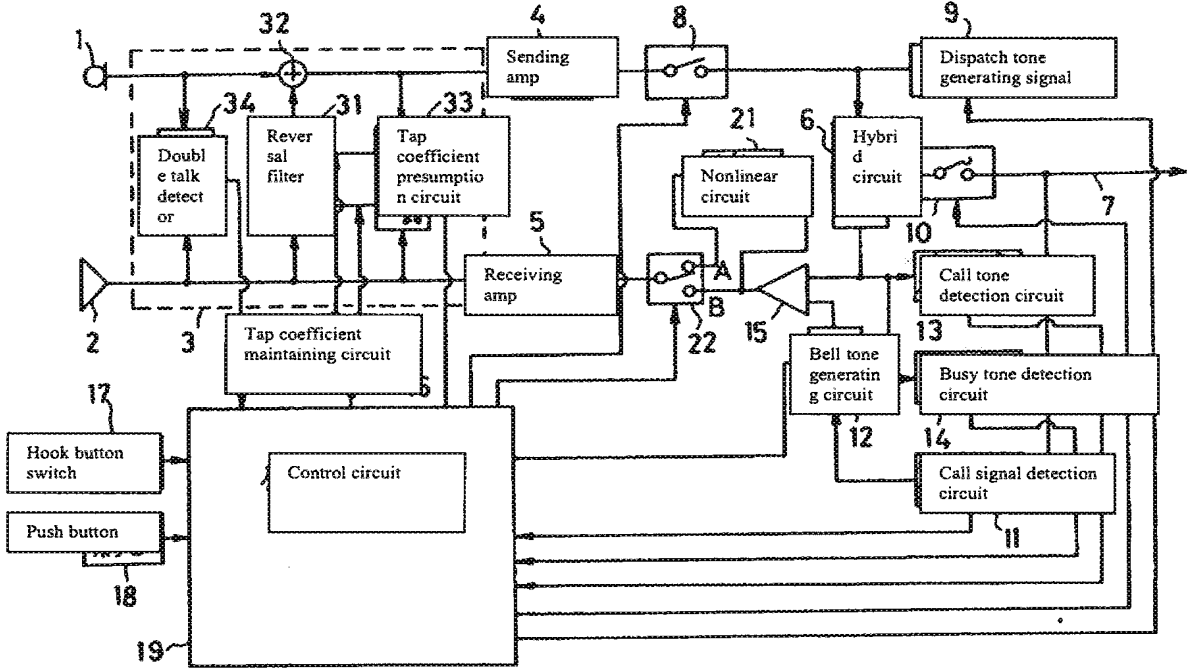


FIG.6

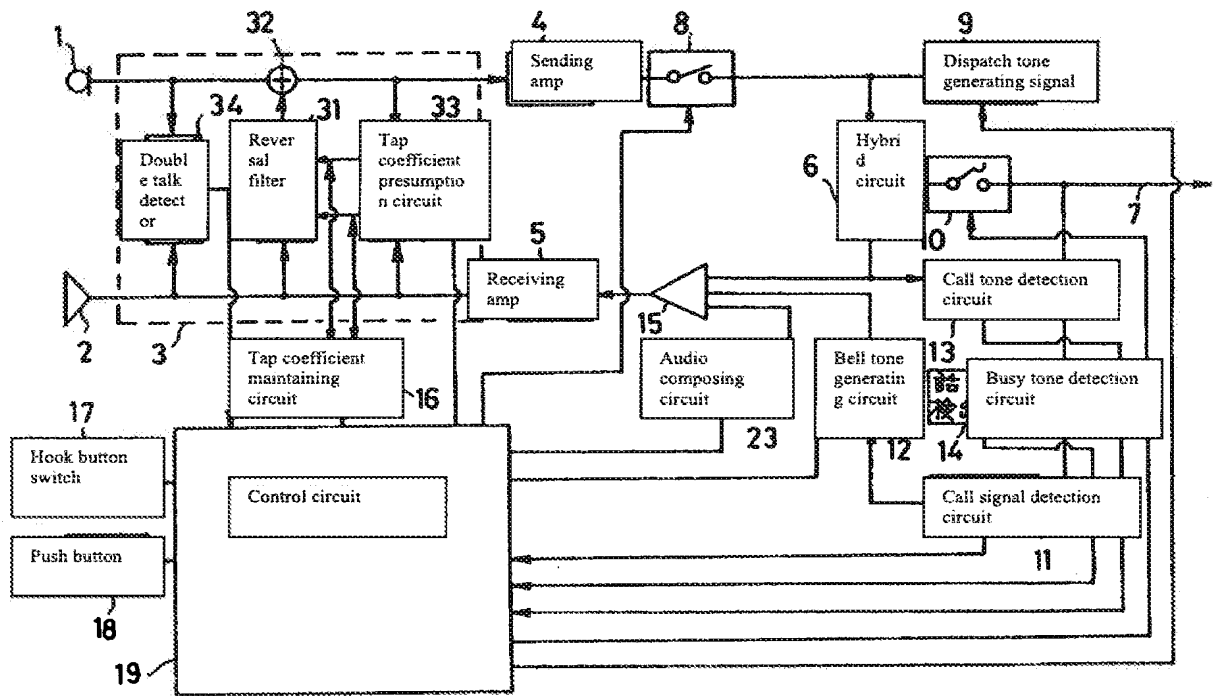


FIG.7

